CLAIMS

1. A method of suppressing noise in a signal, said method comprising the steps of:

estimating a signal to noise ratio for said signal;

classifying said signal to a classification;

calculating a gain for said signal using said signal to noise ratio and said classification; and

modifying said signal using said gain.

2. The method of claim 1 further comprising a step of estimating a pitch correlation for said signal, wherein said calculating step further uses said pitch correlation.

3. The method of claim 1, wherein said signal is one channel of a plurality of channels of a speech signal.

4. The method of claim 1, wherein said calculating step calculates said gain based on $\gamma_{db} = \mu_g (\sigma''_q - \sigma_{th}) + \gamma_n$, and wherein μ_g is adjusted according to said classification.

5. The method of claim 2, wherein said calculating step calculates said gain based on $\gamma_{db} = \mu_g (\sigma''_q - \sigma_{th}) + \gamma_n$, and wherein μ_g is adjusted according to said classification and said pitch correlation.

6. The method of claim 1, wherein said signal is in a time domain, and said method further comprises a step of converting said signal from said time domain to a frequency time prior to said estimating step.

7. The method of claim 1, wherein said signal is in a frequency domain, and said method further comprising a step of converting said signal from said frequncy domain to a time domain after said modifying step.

~~8.~~ A method of suppressing noise in a signal having a first portion and a second portion, said method comprising the steps of:

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computing a voicing parameter using said first portion;

estimating a signal to noise ratio for said second portion;

calculating a gain for said second portion using said signal to noise ratio and said voicing parameter; and

modifying said signal using said gain.

9. The method of claim 8, wherein said first portion of said signal is ahead of said second portion in a time domain.

10. The method of claim 8, wherein said voicing parameter is computed by a speech coder.

11. The method of claim 8, wherein said voicing parameter is a speech classification of said first portion.

12. The method of claim 8, wherein said voicing parameter is a pitch correlation of said first portion.

13. The method of claim 8, wherein said calculating step calculates said gain based on $\gamma_{db} = \mu_g\,(\sigma''_q - \sigma_{th}) + \gamma_n$, and wherein μ_g is adjusted according to said voicing parameter.

14. The method of claim 8, wherein said calculating step calculates said gain

based on $\gamma_{db} = \mu_g (\sigma''_q - \sigma_{th}) + \gamma_n$, and wherein γ_n is adjusted according to said voicing parameter.

15. The method of claim 8, wherein said signal is in a time domain, and said method further comprises a step of converting said signal from said time domain to a frequency time prior to said estimating step.

16. The method of claim 8, wherein said signal is in a frequency domain, and said method further comprising a step of converting said signal from said frequncy domain to a time domain after said modifying step.

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~~17.~~ A noise suppression system comprising:

a signal to noise ratio estimator;

a signal classifier;

a signal gain calculator; and

a signal modifier;

wherein said estimator estimates a signal to noise ratio of said signal, said signal is given a classification using said signal classifier, said signal gain is calculated based on said signal to noise ratio and said classification using said calculator, and wherein said signal modifier modifies said signal by applying said gain.

18. The system of claim 17 further comprising a signal pitch estimator for estimating a pitch correlation of said signal for use by said gain calculator.

19. The system of claim 17 further comprising a frequency-to-time converter to convert said signal from a frequency domain to a time domain.

~~20.~~ A system capable of suppressing noise in a signal having a first portion and

a second portion, said system comprising:

a signal processing module for computing a voicing parameter of said first portion;

a signal to noise ratio estimator;

a signal gain calculator; and

a signal modifier;

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wherein said estimator estimates a signal to noise ratio of said second portion, said second portion gain is calculated based on said signal to noise ratio and said voicing parameter using said calculator, and wherein said signal modifier modifies said second portion by applying said gain.

21. The system of claim 20, wherein said first portion of said signal is ahead of said second portion in a time domain.

22. The system of claim 20, wherein said signal processing module is a speech coder.

23. The system of claim 20, wherein said voicing parameter is a speech classification of said first portion.

24. The system of claim 20, wherein said voicing parameter is a pitch correlation of said first portion.

25. The system of claim 20, wherein said signal gain calculator determines said gain based on $\gamma_{db} = \mu_g (\sigma''_q - \sigma_{th}) + \gamma_n$, and wherein μ_g is adjusted according to said voicing parameter.

26. The system of claim 20 further comprising a frequency-to-time converter to convert said second portion of said signal from a frequency domain to a time domain.